by using the Fourier transform means **18**, the spectrum amplitude suppression means **19** and the inverse Fourier transform means **20** is to remove the quantization noise of the spectrum of the decoded speech decoded by the decoding means **15**. There is quantization noise all over in the decoded speech shown in FIG. **11**(*a*) since the quantization noise is produced in the coding at the speech coding apparatus. Though the part Z of FIG. **11**(*b*),(*c*) are-slightly perceived or masked perceptually, there is quantization noise. There is the case of such quantization noise makes the quality of the decoded speech insufficient. Accordingly, it is possible to prevent the quality of the decoded speech from getting bad by removing the quantization noise in the part which is not perceivable. Such quantization noise can be removed by transforming the decoded speech to the frequency spectrum again and suppressing the part which is slightly perceived or masked even after the decoded speech being output.

As mentioned above, it is a feature of this embodiment to implement the transform means, the amplitude suppression means and the inverse transform means. The transform means transforms the synthetic speech into the frequency spectrum at the speech post processor which transforms the frequency spectrum of the speech synthesized by the speech decoding means. When the frequency component concerned is slightly perceived or masked by the effect of the other frequency components around it, the amplitude suppression means suppresses the amplitude of the frequency component concerned of the frequency spectrum output from the transform means. The inverse transform means transforms the frequency spectrum output from the amplitude suppression means into time domain and outputs it outside.

According to this embodiment, there is an effect of reducing the quality deterioration of the decoded speech produced by quantization noise of the frequency spectrum since the frequency components which are slightly perceived or masked perceptually are masked.

Though the speech post processor **17** shown in FIG. **10** is presented in the above embodiment, it is acceptable to process the output speech **5** by using the Fourier transform means **18**, the spectrum amplitude suppression means **19** and the inverse Fourier transform means **20**. The output speech **5** is output from the speech decoding apparatus **2** shown in FIG. **1**. The output speech will result after suppressing the amplitude of the part which can be masked perceptually in the output speech **5**. It is also acceptable to produce the output speech after suppressing the amplitude of the part which can be masked perceptually in the output speech being output from the speech synthesis apparatus (not illustrated).

What is claimed is:

1. A speech coding apparatus for coding an input speech signal within an analysis window of an analysis frame, comprising:

(a) window locating means for defining a plurality of analysis windows at different locations in the analysis frame, the window locating means including

means for receiving a segment of the input speech signal within each of the analysis windows,

means for calculating a predefined feature of the segment of the input speech signal within each analysis window,

means for comparing the calculated features of each analysis window, and

means for selecting an analysis window based on a result of the comparison;

(b) speech analysis means for extracting characteristic parameters of the input speech signal in the selected analysis window selected by the window locating means; and

(c) coding means for receiving the characteristic parameters and for encoding the characteristic parameters.

2. The speech coding apparatus of claim **1**, wherein the predefined feature is a power of the input speech signal, and wherein an analysis window having a maximum power value is the window selected.

3. The speech coding apparatus of claim **1** or **2**, wherein the speech analysis means comprises:

means for providing a second analysis window different from the selected analysis window: and

means for calculating a value of power of the input speech signal within the second analysis window and for outputting the calculated power value to the coding means.

4. The speech coding apparatus of claim **3**, wherein a center of the second analysis window is placed at a center of the analysis frame.

5. The speech coding apparatus of claim **3**, wherein the analysis frame has a fixed frame length and the second analysis window has a window length which is substantially the same as the analysis frame length.

6. The speech coding apparatus of claim **1**, wherein the selected analysis window is the window having a center which is substantially in the center of the analysis frame.

7. The speech coding apparatus of claim **1**, wherein the analysis frame has a fixed length and the analysis window has a window length which is substantially the same as the frame length.

8. The speech coding apparatus of claim **1**, wherein the predefined feature is a spectrum of the input speech signal of each analysis window and wherein the comparison is a comparison of the spectrums of each analysis window.

9. The speech coding apparatus of claim **1**, wherein the predefined feature is an auto correlation of the input speech signal within each analysis window and wherein an analysis window having an auto correlation function having periodicity is the window selected.

10. A speech coding method for encoding input speech within a selected analysis window of an analysis frame, comprising the steps of:

(a) creating an analysis window having a location in the analysis frame;

(b) calculating a value of power of a segment of the input speech signal within the analysis window;

(c) repeating the above steps, such that a plurality of analysis windows are created at a plurality of different locations within the analysis frame;

(d) comparing the power values of each of the plurality of analysis windows and selecting an analysis window of the plurality of analysis windows having a maximum power value.

11. The speech coding method of claim **10**, further comprising the steps of:

(a) extracting characteristic parameters of the input speech signal within the selected analysis window;

(b) creating a second analysis window and calculating a value of power of the input speech within the second analysis window; and

(c) encoding the extracted characteristic parameters and the calculated power.

12. The speech coding method of claim **11**, wherein the step of creating a second analysis window includes a step of positioning a center of the second analysis window at a center of the analysis frame.